WHAT IS CLAIMED IS:

1. A method of manipulating a digitized sound signal transferred over a packet switched network in the form of data packets, the method being operable in a communication system at a receiver end arranged to decode received sound data packets into sound signal frames to be played back, the method including the step of:

manipulating the length of received signal frames by performing time expansion or time compression of one or more signal frames at time varying intervals and with time varying lengths of the expansion or the compression, said intervals and said lengths being determined so as to maintain a continuous flow of signal samples to be played back.

2. The method of claim 1, wherein each of said time varying lengths is dependent upon a signal fitting criteria with respect to the signal characteristics of the digitized sound signal part to be manipulated.

3. The method of claim 1, wherein a resolution of the length manipulation is a fraction of the time between two samples of said digitized sound signal, thereby enabling an improved signal fitting quality when performing said time expansion or said time compression.

4. The method of claim 1, wherein the receiver end includes a jitter buffer, and including the steps of

storing received data packets to be decoded into signal frames, and

monitoring the jitter buffer to initiate the manipulating step when the timing of the jitter buffer needs to be recovered.

5. The method of claim 4, wherein the time expansion of one or more signal frames is performed for a trailing part of a currently played signal frame if the monitoring of the jitter buffer indicates near or actual buffer underflow, and the manipulating step including repeated time expansions to restore the jitter buffer to its normal working condition.

6. The method of claim 5, wherein said time expansion of said trailing part will constitute a substitution frame if the monitoring of the jitter buffer indicates that a next signal frame, which under normal conditions should follow the currently played signal frame, is not available or deemed not to have been received in due time, thereby providing a lost frame substitution for said next signal frame, after which the time expanded currently played signal frame is merged with a received future signal frame, the length of the time expansion, and thus the length of the substitution frame, being chosen in such way that a smooth transition to said future signal frame can be made.

7. The method of claim 6, wherein said time expansion includes time expanding a heading part of said future signal frame before merging the two frames, thereby improving the lost frame substitution.

8. The method of claim 4, including initiating time compression by real-time statistics from the jitter buffer when two consecutive data packets are available in the jitter buffer, wherein a measure on a smooth transition between two consecutive signal frames, when merging the two signal frames, controls the length of a resulting compressed signal frame.

9. The method of claim 8, wherein said compressed signal frame is merged with yet another consecutive signal frame in the same manner as said merging of said two frames.

10. The method of claim 8, wherein said merging of two signal frames involves merging two signal segments, a trailing segment of one frame with a heading segment of the other frame, by overlap-add, wherein a time-shift of the frame with the heading segment is employed for optimizing the matching of the overlapping part of the two segments.

11. The method of claim 10, wherein said time-shift of the frame with the heading segment has a resolution of a fraction of a time between two samples.

12. The method of claim 11, wherein said heading segment is multiplied with a suitable gain to further optimize the matching with said trailing segment

at the overlapping part, after which a smooth transition back to unity gain is performed in order to avoid sound signal discontinuities.

13. The method of claim 1, wherein use is made of an oscillator model for extracting signal segments used when manipulating the lengths of said received signal frames, the oscillator model including a codebook in which vectors of samples forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

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14. The method of claim 13, wherein said time expansion of a signal frame is performed by matching a true state of a trailing part of such signal frame with said states in said codebook, and reading out a signal segment from said codebook that corresponds to the state having been matched with said true state.

15. The method of claim 13, wherein said signal segments of said codebook have variable lengths, each signal segment forming a trailing part of a signal frame, thereby enabling continuous transition from the time expanded signal frame to a consecutive signal frame.

16. The method of claim 13 wherein time delays between said states in said codebook are incremental delays with a resolution of a fraction of a time between two samples.

17. The method of claim 14, wherein the states and the corresponding segments of said codebook are scaled in order to improve the matching with said true state.

18. The method of claim 14, wherein merging of said true state is performed with the matching state of said codebook.

19. The method of claim 14, wherein said time expansion additionally involves performing the corresponding operations with respect to a heading part of a signal frame being consecutive to the time expanded signal frame.

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20. The method of claim 1, wherein said signal frame, which length is to be manipulated, is either a sound signal frame resulting from a complete decoding

operation of a data packet, or an intermediate time-domain signal frame resulting from a partial decoding operation of a data packet.

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21. The method of claim 1, including the step of using an oscillator model, which oscillator model includes a codebook in which vectors of samples of a received digitized sound signal forms different states, or entries, in the codebook, the codebook storing a corresponding signal segment for each state.

22. The method of claim 1, including storing, in a program storage device, a sequence of instructions for a processor unit for performing the method of claim 1.

23. Apparatus for receiving a digitized sound signal from a packet switched network, the arrangement including:

a memory element for storing a computer program and vectors of samples of a received digitized sound signal together with corresponding signal segments; and

a processor unit for executing the computer program to decode the received sound signal and produce therefrom sound signal frames to be played back by manipulating the length of received signal frames by performing time expansion or time compression of one or more signal frames at time varying intervals and with time varying lengths of the expansion or the compression, said intervals and said lengths being determined so as to maintain a continuous flow of signal samples to be played back.

24. An article of manufacture including a computer memory wherein is located a computer program for causing digitized sound signal transferred over a packet switched network in the form of data packets to be received by a receiver unit of a communication system and to decode received sound data packets into sound signal frames to be played back by manipulating the length of received signal frames by performing time expansion or time compression of one or more signal frames at time varying intervals and with time varying lengths of the expansion or the compression, said intervals and said lengths being determined so as to maintain a continuous flow of signal samples to be played back.

25. A receiver unit for receiving digitized sound in the form of data packets over a packet switched network, the receiver including a processing element having a memory wherein is located a computer program for causing said receiver unit to decode received sound data packets into sound signal frames to be played back by manipulating the length of received signal frames by performing time expansion or time compression of one or more signal frames at time varying intervals and with time varying lengths of the expansion or the compression, said intervals and said lengths being determined so as to maintain a continuous flow of signal samples to be played back.

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